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U.S DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE **TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371**		ATTORNEY'S DOCKET NUMBER 1454.1126 09/980400
INTERNATIONAL APPLICATION NO. PCT/DE00/01662	INTERNATIONAL FILING DATE 24 May 2000	PRIORITY DATE CLAIMED 01 June 1999

TITLE OF INVENTION
METHOD AND ARRANGEMENT FOR SPEECH CODING, USING PHONETIC DECODING AND THE TRANSMISSION OF SPEECH CHARACTERISTICS

APPLICANT(S) FOR DO/EO/US
Christoph ROHE

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a FIRST submission of items concerning a filing under 35 U.S.C. 371.
2. ☒ This is an express request to immediately begin national examination procedures (35 U.S.C. 371(f)).
3. ☒ The US has been elected by the expiration of 19 months from the priority date (PCT Article 31).
4. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2))
   a. ☒ is transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ has been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
5. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
6. ☐ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3))
   a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ have been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US)
7. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
8. ☐ An oath or declaration of the inventor (35 U.S.C. 371(c)(4)).
9. ☐ A translation of the Annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

Items 10-15 below concern document(s) or information included:

10. ☒ An Information Disclosure Statement Under 37 CFR 1.97 and 1.98.
11. ☐ An assignment document for recording.
    Please mail the recorded assignment document to:
    a. ☐ the person whose signature, name & address appears at the bottom of this document.
    b. ☐ the following:
12. ☒ A preliminary amendment.
13. ☒ A substitute specification
14. ☐ A change of power of attorney and/or address letter.
15. ☒ Other items or information:

PCT EASY forms filed with International Application, copy of cover page of International Application as published, International Search Report, and International Preliminary Examination Report.

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☒ The US National Fee (35 U.S.C. 371(c)(1)) and other fees as follows:

CLAIMS	(1) FOR	(2) NUMBER FILED	(3) NUMBER EXTRA	(4) RATE	(5) CALCULATIONS
	TOTAL CLAIMS	15 -20=	0	x $ 18.00	0.00
	INDEPENDENT CLAIMS	2 -3=	0	x $ 84.00	0.00
	MULTIPLE DEPENDENT CLAIM(S) (if applicable)			+$280.00	0.00
	BASIC NATIONAL FEE (37 CFR 1.492(a)(1)-(4): ☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO ......$1,040 ☒ International preliminary examination fee (37 C.F.R. 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO.........$ 890 ☐ International preliminary examination fee (37 C.F.R. 1.482) not paid to USPTO but international search fee (37 C.F.R. 1.445(a)(2) paid to USPTO...$ 740 ☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) but all claims did not satisfy provision of PCT Article 33(1)-(4)...........$ 710 ☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(2) to (4) ....$ 100				890.00
	Surcharge of $130 for furnishing the National fee or oath or declaration later than ☐ 20 ☐ 30 mos. from the earliest claimed priority date (37 CFR 1.482(e)).				0.00
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			SUBTOTAL		0.00
	Processing fee of $130 for furnishing the English Translation later than [ ] 20 [ ] 30 mos. from the earliest claimed priority date (37 CFR 1.482(f)).				
			TOTAL NATIONAL FEE		0.00
	Fee for recording the enclosed assignment (37 CFR 1.21(h)).				+
			TOTAL FEES ENCLOSED		0.00

a. ☐ A check in the amount of $ to cover the above fees is enclosed.

b. ☐ Please charge my Deposit Account No. 19-3935 in the Amount of $ to cover the above fees. A duplicate copy of this sheet is enclosed.

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PATENT TRADEMARK OFFICE

SUBMITTED BY: STAAS & HALSEY LLP			
Type Name	Richard A. Gollhofer	Reg. No.	31,106
Signature	Richard A. Gollhofer	Date	12/3/01

09/980400

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Docket No.: 1454.1126

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of:

Christoph ROHE

Serial No. Group Art Unit:

Confirmation No.

Filed: (concurrently) Examiner:

For: SYSTEM TRANSMITTING DATA SIGNALS WHICH HAVE INDIVIDUAL FEATURES, IN PARTICULAR VOICE SIGNALS (as amended)

PRELIMINARY AMENDMENT

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

Before examination of the above-identified application, please amend the application as follows:

IN THE TITLE:

Please DELETE the Title in its entirety and REPLACE with the following new Title.

-- SYSTEM TRANSMITTING DATA SIGNALS WHICH HAVE INDIVIDUAL FEATURES, IN PARTICULAR VOICE SIGNALS --

IN THE ABSTRACT:

Please DELETE the Abstract in its entirety and substitute the attached new Abstract.

IN THE SPECIFICATION:

Please REPLACE the Specification in its entirety with the attached Substitute Specification.

IN THE CLAIMS:

Please CANCEL claims 1-15 without prejudice or disclaimer and ADD new claims 16-30 in accordance with the following:

16. (NEW) A method for transmitting data signals which have individual features, from a transmitter to a receiver, comprising:

separating the individual features from the data signals at the transmitter to acquire individualization data and free the data signals of the individual features;

transmitting standardized and compressed data signals separately from the individualization data via separate logic channels; and

receiving and processing the standardized and compressed data signals and the individualization data at the receiver to recover the data signals which have individual features.

17. (NEW) The method as claimed in claim 16,

wherein said separating includes converting at least one voice signal into characters at the transmitter, and

wherein said processing includes performing voice synthesis from the characters at the receiver.

18. (NEW) The method as claimed in claim 17, wherein said processing uses an individual-feature knowledge base in which the individualization data is stored in assignment to the characters.

19. (NEW) The method as claimed in claim 18, further comprising transmitting data records for supplementing the individual-feature knowledge base between the transmitter and receiver.

20. (NEW) The method as claimed in claim 19, wherein said transmitting of the data records is controlled with lower priority than said transmitting of the standardized data signals, using pauses during transmission of the standardized data signals.

21. (NEW) The method as claimed in 16, wherein said separating at the transmitter includes transformation into an n-dimensional state space.

22. (NEW) A system for transmitting data signals which have individual features, comprising:

a transmitter to separate the individual features from the data signals to acquire individualization data and standardized data freed of the individual features, to compress the

standardized data to obtain standardized and compressed data signals, and to separately transmit the individualization data and the standardized and compressed data signals, via separate logic channels; and

a receiver to receive the individualization data and the standardized and compressed data signals transmitted from said transmitter, to decompress the standardized and compressed data signals, and to process the standardized data signals and the individualization data to recover the data signals which have the individual features.

23. (NEW) The system as claimed in claim 22, wherein said transmitter comprises

a coder/decoder unit, having an output;

a delay stage connected in parallel with said coder/decoder unit and having an output; and

a difference signal acquisition unit connected to the outputs of said coder/decoder unit and said delay stage.

24. (NEW) The system as claimed in claim 23,

wherein said transmitter further comprises a voice recognition unit to convert voice signals into characters, and

said receiver comprises a voice synthesis unit to synthesize voice signals from the characters.

25. (NEW) The system as claimed in claim 23,

wherein said receiver comprises a first decoder having a function, and

wherein said coder/decoder unit includes a second decoder having a function identical to the function of said first decoder .

26. (NEW) The system as claimed in claim 25, wherein said delay stage has a delay time controllable in real time with adaptation to a current transmitter signal processing delay.

27. (NEW) The system as claimed in claim 26,

wherein said transmitter comprises separation means for separating the standardized and individualization data at an output thereof,

wherein said receiver comprises voice synthesis means for synthesizing voice signals based on the standardized data signals and the individualization data received at an input thereof, and

wherein at least one of said transmitter and said receiver further comprises an individual-feature knowledge base, coupled to the output of said separation means and to the input of said voice synthesis means, for storing individualization data in assignment to the characters.

28. (NEW) The system as claimed in claim 27,

wherein said transmitter comprises a first individualization knowledge base, and

wherein said receiver comprises:

a second individualization knowledge base; and

control means for transmitting new data records from the first individualization knowledge base to supplement contents of the second individualization knowledge base.

29. (NEW) The system as claimed in claim 28, wherein said control means controls transmission of the new data records with a lower priority than transmission of the standardized data signals, using pauses during transmission of the standardized data signals.

30. (NEW) The system as claimed in claim 29, wherein said difference signal acquisition unit comprises means for transforming one of the standardized data signals and the data signals which have individual features into an n-dimensional state space.

REMARKS

This Preliminary Amendment is submitted to improve the form of the English translation as filed. It is respectfully requested that this Preliminary Amendment be entered in the above-referenced application.

In accordance with the foregoing, claims 1-15 have been canceled and claims 16-30 have been added. Thus, claims 16-30 are pending and under consideration.

A substitute specification is also being filed herewith. The substitute specification is accompanied by a marked-up copy of the original specification.

If there are any additional fees associated with filing of this Preliminary Amendment, please charge the same to our Deposit Account No. 19-3935.

Respectfully submitted,

STAAS & HALSEY LLP

Date: 12/3/01

By: Richard A. Gollhofer
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//PRTS

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Description

Method and arrangement for transmitting data signals which have individual features, in particular voice signals

The invention relates to a method for transmitting data signals which have individual features, in particular voice signals, as claimed in the preamble of claim 1, and an arrangement for transmitting data signals which have individual features, in particular voice signals, as claimed in the preamble of claim 7.

The transmission of voice is one of the most important, if not still the most important, telecommunications service. Specifically in the case of mobile communications, on the one hand the limited resources result in the requirement to use as few transmission rates as possible and, on the other hand, the greatly fluctuating transmission properties which are generally significantly worse than those of a line-bound transmission result in relatively high fault rates.

In the course of the development of mobile voice communications, from the start a significant development goal has therefore been to reduce the data rate while simultaneously providing a large degree of resistance to relatively high fault rates. In general terms, a data reduction can take place on the basis of two different procedures: the reduction of redundancy and the reduction of irrelevance. The reduction of redundancy eliminates signal contents which are redundant before the transmission and whose identification is based on the prior knowledge of specific (for example statistical) parameters of the signal. If these redundant signal components are impressed on the signal again after the transmission, there is no transmission-related loss of quality.

In

the reduction of irrelevance, signal components which are considered to be irrelevant to the receiver

are removed before the transmission. If, in this context, the possibility of dispensing with the re-impressing of these signal components after the transmission is selected, there are objective differences between the voice signal generated at the receiver end and the original voice signal, but these are accepted or (at best) cannot be perceived by the human ear.

In the course of the dramatic expansion of mobile communications, the requirements made of the quality of voice transmission have also been increasing. On the other hand, at the same time the fundamental problem of the limited availability of channel resources has been worsening. The development of ever better methods and arrangements for data reduction and compression in the transmission of voice therefore continues to be a development task which is very important at present.

Known digital voice coders are based either on the principle of signal shape coding ("waveform encoding"), in which the analog voice signal is digitized at the transmitter end, and converted with as few faults as possible into an analog signal at the receiver end, and in which an acceptable voice quality is achieved with bit rates of approximately 16 kbit/s to 64 kbit/s, or on the principle of parametric representation (Vocoder principle) in which a level of voice quality which is generally satisfactory only to a limited degree is achieved with a significantly reduced bit rate (to between 400 bit/s and 5 kbits/s). During the latter method, the voice signal is segmented into small sections, during which the voice signal changes only insignificantly and can be characterized by specific excitation or filter parameters. It is not the actual signal but rather the sequence of excitation or filter parameters which is transmitted here. Individual features of the voice (intonation, accents and sentence

melody) can be transmitted with this method only to a very limited degree.

Relatively poor, unnaturally sounding voice transmission with Vocoders has given an impetus to the development of what are referred to as

hybrid coders in which part of the voice frequency band (preferably the low frequency range) is transmitted by means of signal shape coding and the remaining range is transmitted on the basis of the Vocoder principle. This permits a somewhat improved voice quality at the price of a significantly higher transmission rate.

The object of the invention is therefore to disclose an improved method of the generic type and a corresponding arrangement with which a high-quality voice transmission which largely takes into account the individuality of the voice is possible, and it is to be possible to achieve a particularly low transmission rate.

This object is achieved by means of a method having the features of claim 1 and - in terms of its device aspect - by means of an arrangement having the features of claim 7.

The invention includes the fundamental technical idea of separating individual features from the entirety of the data signals at the tranmitter end and separately transmitting the remaining standardized (and compressed) data signals, on the one hand, and the individualization data corresponding to the individual features, on the other. This separate transmission can also take place at different times or else essentially simultaneously as a function of the specific application. In the former case, a knowledge base relating to the individual features can be set up at the receiver end in advance and then the individual features are re-impressed from said knowledge base after the transmission of the standardized data signals. In the latter case, a receiver-end knowledge base relating to the individual features can be dispensed with under certain circumstances.

In one preferred solution, which is to a certain extent located between these two extreme cases, a receiver-end

knowledge base relating to the individual features is built up successively in the course of the transmission, specifically in particular during access to a corresponding transmitter-end knowledge base. This access to the transmitter-end knowledge base is controlled in a preferred embodiment in such a way that the prioritized transmission of the standardized, compressed data signals as main information carrier is not disrupted - i.e. during voice transmission in particular in pauses during speaking or sections of marked word expansion or in times in which a relatively high channel bandwidth is available. In one expedient embodiment, the individual features are separated from the entirety of the data signal in a coder/decoder unit whose decoder part corresponds to the decoder provided at the receiver end for the data signal which is coded at the transmitter end, in a delay stage which is connected in parallel with this unit and in a unit which is connected to the outputs of the two components and has the purpose of acquiring a suitably structured difference signal between the entirety of the data signal present at the transmitter input and the standardized data signal after the Codec has been passed through. In a very simple embodiment, which can, however, of course lead to reproduction of the individual voice only to a specific degree of approximation, a small number of individual features can be viewed as being relatively independent of one another and considered in isolation during a difference formation process. However, a transformation of the signal into an n-dimensional state space (vector space) in which n individual features can be analyzed resolved vectorially is preferred.

In the case - on which our considerations center here - of a data signal which is present as a voice signal, the transmitter has voice recognition means which are known per se, for converting voice into the data

signals in the form of characters, and the receiver has voice synthesis means for synthesizing voice from the characters which can be output acoustically. It is, however, to be noted that the proposed arrangement is suitable not only for voice communication but also basically for any

transmission of signals which have individual features, for example also for the compressed transmission of handwriting or of images with artistic "handwriting" (paintings, graphics etc.).

The delay time of the delay stage which is provided in the preferred embodiment of the separation means can preferably be controlled by adaptation to the current delay which is brought about by the transmitter-end signal processing, i.e. the coding/decoding and if appropriate voice analysis. This can provide a considerable overall saving in time during the processing of signals because the assumption of a fixed signal delay for the processing operations which are associated with the separation of the individual features would have to be adapted to the "worst case" of a data signal sequence, which requires processing which requires a maximum amount of time.

As a function of the specific application, in particular also of the memory and processing capacities which are available at the transmitter and/or receiver end, the transmitter and/or the receiver has an individual-feature knowledge base which is connected at the input end to the separation means and at the output end at least indirectly to an input of the voice synthesis means, in order to store individual features in assignment to the associated characters as a representation of the standardized data signals. In one particularly advantageous embodiment, the transmitter receives here a first such knowledge base and the receiver a second such knowledge base, and control means for transmitting new data records for supplementing the memory contents of the second knowledge base from the stock of the first are provided, said control means ensuring efficient and reliable transmission of the corresponding individualization data via the separate channel. Its

transmission is carried out in particular with lower priority than the standardized data signals, in particular in pauses in the transmission of the latter.

Advantages and expediencies of the invention emerge also from the subclaims and from the following description of a preferred embodiment with reference to the figure. The latter shows a transmission arrangement 1 with a transmitter 3 and a receiver 5, which can be constructed, for example, as a transmitter part or a receiver part of a mobile radio transmission link. The figure shows only the components which are essential for the explanation of the invention, while the other components of a mobile radio transmitter or receiver part are omitted here for the sake of clarity.

The sound waves (input voice signals) which are picked up by a microphone 7 are digitized in an A/D converter 9, if appropriate after preprocessing which includes amplification and/or filtering in order to suppress interference, and the signal path is divided into two sub-paths at a node 11 at the output of the A/D converter 9. In a first sub-path 13a, the digitized voice signal is firstly subjected to a voice recognition algorithm (known per se) and a voice recognition stage 15, the input voice signals being converted into characters, and then subjected to coding in a coder 17. The characters which are generated in the voice recognition stage can be letters, for example in ASCII code or else numbers for words or syllables. Depending on their probability of occurrence, these characters are converted into codes of differing length according to the principle of high probability = short code, low probability = long code. For each of the previously formed characters, the code which is particularly used depends on the predecessors of the code because the probability of a syllable or of a word and also of the intonation of the syllable or word depends on the previous one. The coded characters are conditioned in a transmission stage known per se and therefore not illustrated in the figure, and transmitted via a first logic channel CH1 to the receiver where they are firstly preprocessed

in terms of RF, and if appropriate also in accordance with the specifications of a particular mobile radio protocol by despreading, descrambling or the like in a receiver stage, which is also known and therefore omitted here, and then fed to a voice decoder 19. The further receiver-end signal processing is described further below.

The coded voice data which is made available at the output of the transmitter-end coder 17 is transmitted not only to the receiver but also immediately subjected to decoding again in a transmitter-end voice decoder 21 which functionally corresponds completely to the receiver-end voice decoder 19. A transformation into an n-dimensional state space is then carried out in a first transformation stage 23a by means of algorithms which are known per se. The signal which is branched at the node 11 is also subjected to a corresponding transformation in the second sub-signal path 13b in a second transformation stage 23b after said signal has been subjected to a delay, synchronized to the signal delay in the first sub-signal path 13a, in a delay stage 25. It is to be noted here that the data signal which is present at the input of the first transformation stage is a standardized voice signal which is reduced by the process of coding and subsequent decoding in the stages 17, 21, while the data signal which is present at the input of the second transformation stage 23 is still the entirety of the data signal which is merely suitably delayed. The delay time which is impressed by the delay stage 25 is controlled in adaptation to the delay in the processing chain of the stages 15, 17 and 21; in the figure it is assumed (in a somewhat simplifying fashion) that control is carried out as a function of the result of the voice recognition, i.e. on the basis of the voice recognition stage 15.

The description in the n-dimensional state space makes it possible for the individual features of the voice to be obtained in isolation as a result of the difference formation in the subtraction stage 27. Owing to the precise synchronization of the two

sub-signal paths 13a, 13b, these individual features can be unambiguously assigned to the characters obtained as a result of voice recognition, and can be stored in this assignment in a transmitter-end individual-feature knowledge base 29.

The individualization data which represents the individual features is transmitted to the receiver 5 via a separate, second logic channel CH2 at whose beginning and end there is in each case a Codec 31, 33. This data is firstly tested in the receiver using a specific control channel CH3 in a comparator and memory control stage (not shown in the figure) to determine whether or not it is already contained in a receiver-end individual-feature knowledge base 35. If this is not the case, it is stored in the receiver-end knowledge base 35 - again in assignment to the corresponding characters of the standardized voice signals (transmitted separately to the receiver 5). The receiver-end knowledge base 35 is thus to a certain degree "harmonized" with the transmitter-end knowledge base 29 in terms of its data stock so that only information relating to the individual features which is not already present in the receiver-end knowledge base 35 has to be transmitted via the separate channel. The quantity of data which is to be transmitted here can therefore be kept comparatively small.

Because the individualization data also has virtually no indispensable information value, it is transmitted with lower priority than the standardized voice signals. A transmission of this data can therefore be carried out, for example, only in the pauses during speaking or in time intervals of high word expansion. The proposed solution therefore requires a separate logic data channel but does not require any additional channel resources. The separate transmission of the individualization data from the standardized voice data

in the context of the provision of a "simultaneously learning" knowledge base for the individualization data at least in the receiver (preferably in the transmitter and receiver) even

makes possible a large reduction in the transmission bandwidth required despite the fact that a relatively high voice quality is achieved (dependent of course on the expenditure on the processing of the individualization features).

The individualization data which is available in the receiver-end individual-feature knowledge base 35 - again synchronized with the standardized voice signal data leaving the decoder 19 - is fed to a voice synthesis unit ("voice generator") 37 where the standardized voice signals are combined with the individualization data and an acoustic output unit 39 for converting sound is connected to its output. The method of operation of the voice synthesis unit 37 and of the output unit 39 are known per se and are therefore not explained in more detail here; a particular feature of the voice synthesis unit 37 is, however, the additional input for the individualization data and the implementation of an algorithm which is suitable for combining this individualization data with the standardized voice signals.

The embodiment of the invention is not restricted to the example outlined here but is also possible in a multiplicity of refinements and specific applications, a number of which have been already mentioned further above.

Patent Claims

1. A method for transmitting data signals which have individual features, in particular voice signals, from a transmitter (3), which has compression means (17) for compressing the data signals, to a receiver (5) which has decompression means (19) for decompressing the data signals compressed at the transmitter end, characterized in that at the transmitter end the individual features are separated from the data signals in order to acquire individualization data and the data signals which are freed of the individual features, standardized and compressed, on the one hand, and the individualization data, on the other, are transmitted separately via separate logic channels (CH1, CH2) and the separately received standardized data signals and individualization data are processed at the receiver end in order to recover the data signals which have individual features.

2. The method as claimed in claim 1, characterized in that voice is converted into data signals in the form of characters at the transmitter end and voice synthesis from the characters is performed at the receiver end.

3. The method as claimed in claim 1 or 2, characterized in that the receiver-end processing is carried out using data from an individual-feature knowledge base (29, 35) in which individualization data is stored in assignment to the characters.

4. The method as claimed in claim 3, characterized in that data records for supplementing the receiver-end individual-feature knowledge base (35) are

transmitted between the transmitter (3) and receiver (5).

5. The method as claimed in claim 4, characterized in that the transmission of the data records is controlled with lower priority than the transmission of the standardized data signals, in particular in pauses during their transmission.

6. The method as claimed in one of the preceding claims, characterized in that a transformation into an n-dimensional state space is performed at the transmission end in order to separate the individualization data from the standardized data signals.

7. An arrangement (1) for transmitting data signals which have individual features, in particular voice signals, with a transmitter (3) which has compression means (17) for compressing the data signals, and a receiver (5) which has decompression means (19) for decompressing the data signals which are compressed at the transmitter end, characterized in that the transmitter has separation means (15 to 27) for separating the individual features from the data signals in order to acquire individualization data and means for the separate transmission of the standardized and compressed data signals which are freed of the individual features, on the one hand, and of the individualization data, on the other, via separate logic channels (CH1, CH2), and
the receiver has means (19, 35, 37) for processing the separately received, standardized data signals and individualization data for recovering the data signals which have the individual features.

8. The arrangement as claimed in claim 7, characterized in that the separation means (15 to 27) comprise a coder/decoder unit (17, 21) and a delay stage (25) which is connected in parallel therewith, as well as

a difference signal acquisition unit (23a, 23b, 27) which is connected to the outputs of the coder/decoder unit and of the delay stage.

9. The arrangement as claimed in claim 7 or 8, characterized in that the transmitter (3) has voice recognition means (15) for converting voice into the data signals in the form of characters, and the receiver (5) has voice synthesis means (37) for synthesizing voice from the characters.

10. The arrangement as claimed in claim 8 or 9, characterized in that the transmitter-end coder/decoder unit (17, 21) has a decoder (21) which has a function which is identical to a receiver-end decoder (19).

11. The arrangement as claimed in any one of claims 8 to 10, characterized in that a delay time of the delay stage (25) can be controlled in real time with adaptation to the current transmitter-end signal processing delay.

12. The arrangement as claimed in one of claims 9 to 11, characterized in that the transmitter (3) and/or the receiver (5) has an individual-feature knowledge base (29, 35), at least indirectly connected to the output of the separation means (15 to 27) and to an input of the voice synthesis means (37), for storing individualization data in assignment to the characters.

13. The arrangement as claimed in claim 12, characterized in that the transmitter (3) has a first individualization knowledge base (29) and the receiver (5) has a second individualization knowledge base (35) and control means (31, 33, CH3) for transmitting new data records from the first

individualization knowledge base in order to supplement the memory contents of the second individualization knowledge base.

14. The arrangement as claimed in claim 13, characterized in that the control means for controlling the transmission of new data records are constructed with a lower priority than the transmission of the standardized data signals, in particular in pauses during their transmission.

15. The arrangement as claimed in one of claims 8 to 14, characterized in that the difference signal acquisition unit (23a, 23b, 27) has means for transforming the standardized data signals and/or the entirety of the data signal into an n-dimensional state space.

GR 99 P 1965

Abstract

Method and arrangement for transmitting data signals which have individual features, in particular voice signals

Method and arrangement (1) for transmitting data signals which have individual features, in particular voice signals, having a transmitter (3) and a receiver (5), the transmitter having separation means (15 to 27) for separating the individual features from the data signals and means for the separate transmission of the standardized and compressed data signals which are freed of the individual features, on the one hand, and of the individualization data, on the other, via separate logic channels (CH1, CH2).

Figure

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SUBSTITUTE SPECIFICATION

TITLE OF THE INVENTION

SYSTEM FOR TRANSMITTING DATA SIGNALS WHICH HAVE INDIVIDUAL FEATURES, IN PARTICULAR VOICE SIGNALS

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application is based on and hereby claims priority to German Patent Application No. 19925264.5 filed on June 1, 1999, the contents of which are hereby incorporated by reference.

BACKGROUND OF THE INVENTION

1. Field of the Invention

[0002] The invention relates to a method and system for transmitting data signals which have individual features, in particular voice signals.

2. Description of the Related Art

[0003] The transmission of voice is one of the most important, if not still the most important, telecommunications service. Specifically in the case of mobile communications, on the one hand the limited resources result in the requirement to use as few transmission rates as possible and, on the other hand, the greatly fluctuating transmission properties which are generally significantly worse than those of a line-bound transmission result in relatively high fault rates.

[0004] In the course of the development of mobile voice communications, from the start a significant development goal has therefore been to reduce the data rate while simultaneously providing a large degree of resistance to relatively high fault rates. In general terms, a data reduction can take place on the basis of two different procedures: the reduction of redundancy and the reduction of irrelevance. The reduction of redundancy eliminates signal contents which are redundant before the transmission and whose identification is based on the prior knowledge of specific (for example statistical) parameters of the signal. If these redundant signal components are impressed on the signal again after the transmission, there is no transmission-related loss of quality. In the reduction of irrelevance, signal components which are considered to be irrelevant to the receiver are removed before the transmission. If, in this context, the possibility of dispensing with the re-impressing of these signal components after the

transmission is selected, there are objective differences between the voice signal generated at the receiver end and the original voice signal, but these are accepted or (at best) cannot be perceived by the human ear.

[0005] In the course of the dramatic expansion of mobile communications, the requirements made of the quality of voice transmission have also been increasing. On the other hand, at the same time the fundamental problem of the limited availability of channel resources has been worsening. The development of ever better methods and systems for data reduction and compression in the transmission of voice therefore continues to be a development task which is very important at present.

[0006] Known digital voice coders are based either on the principle of signal shape coding ("waveform encoding"), in which the analog voice signal is digitized at the transmitter end, and converted with as few faults as possible into an analog signal at the receiver end, and in which an acceptable voice quality is achieved with bit rates of approximately 16 kbit/s to 64 kbit/s, or on the principle of parametric representation (Vocoder principle) in which a level of voice quality which is generally satisfactory only to a limited degree is achieved with a significantly reduced bit rate (to between 400 bit/s and 5 kbits/s). During the latter method, the voice signal is segmented into small sections, during which the voice signal changes only insignificantly and can be characterized by specific excitation or filter parameters. It is not the actual signal but rather the sequence of excitation or filter parameters which is transmitted here. Individual features of the voice (intonation, accents and sentence melody) can be transmitted with this method only to a very limited degree.

[0007] Relatively poor, unnaturally sounding voice transmission with Vocoders has given an impetus to the development of what are referred to as hybrid coders in which part of the voice frequency band (preferably the low frequency range) is transmitted by signal shape coding and the remaining range is transmitted on the basis of the Vocoder principle. This permits a somewhat improved voice quality at the price of a significantly higher transmission rate.

SUMMARY OF THE INVENTION

[0008] The object of the invention is therefore to disclose an improved method of the generic type and a corresponding system with which a high-quality voice transmission which largely takes into account the individuality of the voice is possible, and it is to be possible to achieve a particularly low transmission rate.

[0009] The invention includes the fundamental technical idea of separating individual features from the entirety of the data signals at the transmitter end and separately transmitting the remaining standardized (and compressed) data signals, on the one hand, and the individualization data corresponding to the individual features, on the other. This separate transmission can also take place at different times or else essentially simultaneously as a function of the specific application. In the former case, a knowledge base relating to the individual features can be set up at the receiver end in advance and then the individual features are re-impressed from said knowledge base after the transmission of the standardized data signals. In the latter case, a receiver-end knowledge base relating to the individual features can be dispensed with under certain circumstances.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010] These and other objects and advantages of the present invention will become more apparent and more readily appreciated from the following description of the preferred embodiments, taken in conjunction with the accompanying drawings of which:

The figure is a block diagram of a system according to the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

[0011] Reference will now be made in detail to the preferred embodiments of the present invention, examples of which are illustrated in the accompanying drawings, wherein like reference numerals refer to like elements throughout.

[0012] In one preferred solution, which is to a certain extent located between two extreme cases, a receiver-end knowledge base relating to the individual features is built up successively in the course of the transmission, specifically in particular during access to a corresponding transmitter-end knowledge base. This access to the transmitter-end knowledge base is controlled in a preferred embodiment in such a way that the prioritized transmission of the standardized, compressed data signals as main information carrier is not disrupted - i.e. during voice transmission in particular in pauses during speaking or sections of marked word expansion or in times in which a relatively high channel bandwidth is available. In one expedient embodiment, the individual features are separated from the entirety of the data signal in a coder/decoder unit whose decoder part corresponds to the decoder provided at the receiver end for the data signal which is coded at the transmitter end, in a delay stage which is connected in parallel with this unit and in a unit which is connected to the outputs of the two components and has the purpose of acquiring a suitably structured difference signal between

the entirety of the data signal present at the transmitter input and the standardized data signal after the Codec has been passed through. In a very simple embodiment, which can, however, of course lead to reproduction of the individual voice only to a specific degree of approximation, a small number of individual features can be viewed as being relatively independent of one another and considered in isolation during a difference formation process. However, a transformation of the signal into an n-dimensional state space (vector space) in which n individual features can be analyzed resolved vectorially is preferred.

[0013] In the case - on which our considerations center here - of a data signal which is present as a voice signal, the transmitter has voice recognition means which are known per se, for converting voice into the data signals in the form of characters, and the receiver has voice synthesis means for synthesizing voice from the characters which can be output acoustically. It is, however, to be noted that the proposed system is suitable not only for voice communication but also basically for any transmission of signals which have individual features, for example also for the compressed transmission of handwriting or of images with artistic "handwriting" (paintings, graphics etc.).

[0014] The delay time of the delay stage which is provided in the preferred embodiment of the separation means can preferably be controlled by adaptation to the current delay which is brought about by the transmitter-end signal processing, i.e. the coding/decoding and if appropriate voice analysis. This can provide a considerable overall saving in time during the processing of signals because the assumption of a fixed signal delay for the processing operations which are associated with the separation of the individual features would have to be adapted to the "worst case" of a data signal sequence, which requires processing which requires a maximum amount of time.

[0015] As a function of the specific application, in particular also of the memory and processing capacities which are available at the transmitter and/or receiver end, the transmitter and/or the receiver has an individual-feature knowledge base which is connected at the input end to the separation means and at the output end at least indirectly to an input of the voice synthesis means, in order to store individual features in assignment to the associated characters as a representation of the standardized data signals. In one particularly advantageous embodiment, the transmitter receives here a first such knowledge base and the receiver a second such knowledge base, and control means for transmitting new data records for supplementing the memory contents of the second knowledge base from the stock of the first are provided, said control means ensuring efficient and reliable transmission of the

corresponding individualization data via the separate channel. Its transmission is carried out in particular with lower priority than the standardized data signals, in particular in pauses in the transmission of the latter.

[0016] The figure shows a transmission system 1 with a transmitter 3 and a receiver 5, which can be constructed, for example, as a transmitter part or a receiver part of a mobile radio transmission link. The figure shows only the components which are essential for the explanation of the invention, while the other components of a mobile radio transmitter or receiver part are omitted here for the sake of clarity.

[0017] The sound waves (input voice signals) which are picked up by a microphone 7 are digitized in an A/D converter 9, if appropriate after preprocessing which includes amplification and/or filtering in order to suppress interference, and the signal path is divided into two sub-paths at a node 11 at the output of the A/D converter 9. In a first sub-path 13a, the digitized voice signal is firstly subjected to a voice recognition algorithm (known per se) and a voice recognition stage 15, the input voice signals being converted into characters, and then subjected to coding in a coder 17. The characters which are generated in the voice recognition stage can be letters, for example in ASCII code or else numbers for words or syllables. Depending on their probability of occurrence, these characters are converted into codes of differing length according to the principle of high probability = short code, low probability = long code. For each of the previously formed characters, the code which is particularly used depends on the predecessors of the code because the probability of a syllable or of a word and also of the intonation of the syllable or word depends on the previous one. The coded characters are conditioned in a transmission stage known per se and therefore not illustrated in the figure, and transmitted via a first logic channel CH1 to the receiver where they are firstly preprocessed in terms of RF, and if appropriate also in accordance with the specifications of a particular mobile radio protocol by despreading, descrambling or the like in a receiver stage, which is also known and therefore omitted here, and then fed to a voice decoder 19. The further receiver-end signal processing is described further below.

[0018] The coded voice data which is made available at the output of the transmitter-end coder 17 is transmitted not only to the receiver but also immediately subjected to decoding again in a transmitter-end voice decoder 21 which functionally corresponds completely to the receiver-end voice decoder 19. A transformation into an n-dimensional state space is then carried out in a first transformation stage 23a by algorithms which are known per se. The signal which is branched at the node 11 is also subjected to a corresponding transformation in the

second sub-signal path 13b in a second transformation stage 23b after said signal has been subjected to a delay, synchronized to the signal delay in the first sub-signal path 13a, in a delay stage 25. It is to be noted here that the data signal which is present at the input of the first transformation stage is a standardized voice signal which is reduced by the process of coding and subsequent decoding in the stages 17, 21, while the data signal which is present at the input of the second transformation stage 23 is still the entirety of the data signal which is merely suitably delayed. The delay time which is impressed by the delay stage 25 is controlled in adaptation to the delay in the processing chain of the stages 15, 17 and 21; in the figure it is assumed (in a somewhat simplifying fashion) that control is carried out as a function of the result of the voice recognition, i.e. on the basis of the voice recognition stage 15.

[0019] The description in the n-dimensional state space makes it possible for the individual features of the voice to be obtained in isolation as a result of the difference formation in the subtraction stage 27. Owing to the precise synchronization of the two sub-signal paths 13a, 13b, these individual features can be unambiguously assigned to the characters obtained as a result of voice recognition, and can be stored in this assignment in a transmitter-end individual-feature knowledge base 29.

[0020] The individualization data which represents the individual features is transmitted to the receiver 5 via a separate, second logic channel CH2 at whose beginning and end there is in each case a Codec 31, 33. This data is firstly tested in the receiver using a specific control channel CH3 in a comparator and memory control stage (not shown in the figure) to determine whether or not it is already contained in a receiver-end individual-feature knowledge base 35. If this is not the case, it is stored in the receiver-end knowledge base 35 - again in assignment to the corresponding characters of the standardized voice signals (transmitted separately to the receiver 5). The receiver-end knowledge base 35 is thus to a certain degree "harmonized" with the transmitter-end knowledge base 29 in terms of its data stock so that only information relating to the individual features which is not already present in the receiver-end knowledge base 35 has to be transmitted via the separate channel. The quantity of data which is to be transmitted here can therefore be kept comparatively small.

[0021] Because the individualization data also has virtually no indispensable information value, it is transmitted with lower priority than the standardized voice signals. A transmission of this data can therefore be carried out, for example, only in the pauses during speaking or in time intervals of high word expansion. The proposed solution therefore requires a separate logic data channel but does not require any additional channel resources. The separate transmission

of the individualization data from the standardized voice data in the context of the provision of a "simultaneously learning" knowledge base for the individualization data at least in the receiver (preferably in the transmitter and receiver) even makes possible a large reduction in the transmission bandwidth required despite the fact that a relatively high voice quality is achieved (dependent of course on the expenditure on the processing of the individualization features).

[0022] The individualization data which is available in the receiver-end individual-feature knowledge base 35 - again synchronized with the standardized voice signal data leaving the decoder 19 - is fed to a voice synthesis unit ("voice generator") 37 where the standardized voice signals are combined with the individualization data and an acoustic output unit 39 for converting sound is connected to its output. The method of operation of the voice synthesis unit 37 and of the output unit 39 are known per se and are therefore not explained in more detail here; a particular feature of the voice synthesis unit 37 is, however, the additional input for the individualization data and the implementation of an algorithm which is suitable for combining this individualization data with the standardized voice signals.

[0023] The embodiment of the invention is not restricted to the example outlined here but is also possible in a multiplicity of refinements and specific applications, a number of which have been already mentioned further above. Thus, it will be understood that variations and modifications can be effected within the spirit and scope of the invention.

SUBSTITUTE ABSTRACT

SYSTEM TRANSMITTING DATA SIGNALS WHICH HAVE INDIVIDUAL FEATURES, IN PARTICULAR VOICE SIGNALS

Data signals are transmitted which have individual features, in particular voice signals by a transmitter and a receiver. The transmitter separates the individual features from the data signals and separately transmits the individualization data and the standardized and compressed data signals which are freed of the individual features, via separate logic channels.

MARKED-UP SUBSTITUTE SPECIFICATION

[Description]

TITLE OF THE INVENTION

[METHOD AND ARRANGEMENT] SYSTEM FOR TRANSMITTING DATA SIGNALS WHICH HAVE INDIVIDUAL FEATURES, IN PARTICULAR VOICE SIGNALS

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application is based on and hereby claims priority to German Patent Application No. 19925264.5 filed on June 1, 1999, the contents of which are hereby incorporated by reference.

BACKGROUND OF THE INVENTION

1. Field of the Invention

[0002] The invention relates to a method and system for transmitting data signals which have individual features, in particular voice signals[, as claimed in the preamble of claim 1, and an arrangement for transmitting data signals which have individual features, in particular voice signals, as claimed in the preamble of claim 7].

2. Description of the Related Art

[0003] The transmission of voice is one of the most important, if not still the most important, telecommunications service. Specifically in the case of mobile communications, on the one hand the limited resources result in the requirement to use as few transmission rates as possible and, on the other hand, the greatly fluctuating transmission properties which are generally significantly worse than those of a line-bound transmission result in relatively high fault rates.

[0004] In the course of the development of mobile voice communications, from the start a significant development goal has therefore been to reduce the data rate while simultaneously providing a large degree of resistance to relatively high fault rates. In general terms, a data reduction can take place on the basis of two different procedures: the reduction of redundancy and the reduction of irrelevance. The reduction of redundancy eliminates signal contents which are redundant before the transmission and whose identification is based on the prior knowledge of specific (for example statistical) parameters of the signal. If these redundant signal components are impressed on the signal again after the transmission, there is no transmission-related loss of quality. In the reduction of irrelevance, signal components which are considered

to be irrelevant to the receiver are removed before the transmission. If, in this context, the possibility of dispensing with the re-impressing of these signal components after the transmission is selected, there are objective differences between the voice signal generated at the receiver end and the original voice signal, but these are accepted or (at best) cannot be perceived by the human ear.

[0005] In the course of the dramatic expansion of mobile communications, the requirements made of the quality of voice transmission have also been increasing. On the other hand, at the same time the fundamental problem of the limited availability of channel resources has been worsening. The development of ever better methods and [arrangements] <u>systems</u> for data reduction and compression in the transmission of voice therefore continues to be a development task which is very important at present.

[0006] Known digital voice coders are based either on the principle of signal shape coding ("waveform encoding"), in which the analog voice signal is digitized at the transmitter end, and converted with as few faults as possible into an analog signal at the receiver end, and in which an acceptable voice quality is achieved with bit rates of approximately 16 kbit/s to 64 kbit/s, or on the principle of parametric representation (Vocoder principle) in which a level of voice quality which is generally satisfactory only to a limited degree is achieved with a significantly reduced bit rate (to between 400 bit/s and 5 kbits/s). During the latter method, the voice signal is segmented into small sections, during which the voice signal changes only insignificantly and can be characterized by specific excitation or filter parameters. It is not the actual signal but rather the sequence of excitation or filter parameters which is transmitted here. Individual features of the voice (intonation, accents and sentence melody) can be transmitted with this method only to a very limited degree.

[0007] Relatively poor, unnaturally sounding voice transmission with Vocoders has given an impetus to the development of what are referred to as hybrid coders in which part of the voice frequency band (preferably the low frequency range) is transmitted by [means of] signal shape coding and the remaining range is transmitted on the basis of the Vocoder principle. This permits a somewhat improved voice quality at the price of a significantly higher transmission rate.

<u>SUMMARY OF THE INVENTION</u>

[0008] The object of the invention is therefore to disclose an improved method of the generic type and a corresponding [arrangement] <u>system</u> with which a high-quality voice transmission

which largely takes into account the individuality of the voice is possible, and it is to be possible to achieve a particularly low transmission rate. [This object is achieved by means of a method having the features of claim 1 and - in terms of its device aspect - by means of an arrangement having the features of claim 7.]

[0009] The invention includes the fundamental technical idea of separating individual features from the entirety of the data signals at the [tranmitter] transmitter end and separately transmitting the remaining standardized (and compressed) data signals, on the one hand, and the individualization data corresponding to the individual features, on the other. This separate transmission can also take place at different times or else essentially simultaneously as a function of the specific application. In the former case, a knowledge base relating to the individual features can be set up at the receiver end in advance and then the individual features are re-impressed from said knowledge base after the transmission of the standardized data signals. In the latter case, a receiver-end knowledge base relating to the individual features can be dispensed with under certain circumstances.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010] These and other objects and advantages of the present invention will become more apparent and more readily appreciated from the following description of the preferred embodiments, taken in conjunction with the accompanying drawings of which:

The figure is a block diagram of a system according to the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

[0011] Reference will now be made in detail to the preferred embodiments of the present invention, examples of which are illustrated in the accompanying drawings, wherein like reference numerals refer to like elements throughout.

[0012] In one preferred solution, which is to a certain extent located between [these] two extreme cases, a receiver-end knowledge base relating to the individual features is built up successively in the course of the transmission, specifically in particular during access to a corresponding transmitter-end knowledge base. This access to the transmitter-end knowledge base is controlled in a preferred embodiment in such a way that the prioritized transmission of the standardized, compressed data signals as main information carrier is not disrupted - i.e. during voice transmission in particular in pauses during speaking or sections of marked word expansion or in times in which a relatively high channel bandwidth is available. In one expedient embodiment, the individual features are separated from the entirety of the data signal

in a coder/decoder unit whose decoder part corresponds to the decoder provided at the receiver end for the data signal which is coded at the transmitter end, in a delay stage which is connected in parallel with this unit and in a unit which is connected to the outputs of the two components and has the purpose of acquiring a suitably structured difference signal between the entirety of the data signal present at the transmitter input and the standardized data signal after the Codec has been passed through. In a very simple embodiment, which can, however, of course lead to reproduction of the individual voice only to a specific degree of approximation, a small number of individual features can be viewed as being relatively independent of one another and considered in isolation during a difference formation process. However, a transformation of the signal into an n-dimensional state space (vector space) in which n individual features can be analyzed resolved vectorially is preferred.

[0013] In the case - on which our considerations center here - of a data signal which is present as a voice signal, the transmitter has voice recognition means which are known per se, for converting voice into the data signals in the form of characters, and the receiver has voice synthesis means for synthesizing voice from the characters which can be output acoustically. It is, however, to be noted that the proposed [arrangement] <u>system</u> is suitable not only for voice communication but also basically for any transmission of signals which have individual features, for example also for the compressed transmission of handwriting or of images with artistic "handwriting" (paintings, graphics etc.).

[0014] The delay time of the delay stage which is provided in the preferred embodiment of the separation means can preferably be controlled by adaptation to the current delay which is brought about by the transmitter-end signal processing, i.e. the coding/decoding and if appropriate voice analysis. This can provide a considerable overall saving in time during the processing of signals because the assumption of a fixed signal delay for the processing operations which are associated with the separation of the individual features would have to be adapted to the "worst case" of a data signal sequence, which requires processing which requires a maximum amount of time.

[0015] As a function of the specific application, in particular also of the memory and processing capacities which are available at the transmitter and/or receiver end, the transmitter and/or the receiver has an individual-feature knowledge base which is connected at the input end to the separation means and at the output end at least indirectly to an input of the voice synthesis means, in order to store individual features in assignment to the associated characters as a representation of the standardized data signals. In one particularly

advantageous embodiment, the transmitter receives here a first such knowledge base and the receiver a second such knowledge base, and control means for transmitting new data records for supplementing the memory contents of the second knowledge base from the stock of the first are provided, said control means ensuring efficient and reliable transmission of the corresponding individualization data via the separate channel. Its transmission is carried out in particular with lower priority than the standardized data signals, in particular in pauses in the transmission of the latter.

[0016] [Advantages and expediencies of the invention emerge also from the subclaims and from the following description of a preferred embodiment with reference to the] The figure[. The latter] shows a transmission [arrangement] system 1 with a transmitter 3 and a receiver 5, which can be constructed, for example, as a transmitter part or a receiver part of a mobile radio transmission link. The figure shows only the components which are essential for the explanation of the invention, while the other components of a mobile radio transmitter or receiver part are omitted here for the sake of clarity.

[0017] The sound waves (input voice signals) which are picked up by a microphone 7 are digitized in an A/D converter 9, if appropriate after preprocessing which includes amplification and/or filtering in order to suppress interference, and the signal path is divided into two sub-paths at a node 11 at the output of the A/D converter 9. In a first sub-path 13a, the digitized voice signal is firstly subjected to a voice recognition algorithm (known per se) and a voice recognition stage 15, the input voice signals being converted into characters, and then subjected to coding in a coder 17. The characters which are generated in the voice recognition stage can be letters, for example in ASCII code or else numbers for words or syllables. Depending on their probability of occurrence, these characters are converted into codes of differing length according to the principle of high probability = short code, low probability = long code. For each of the previously formed characters, the code which is particularly used depends on the predecessors of the code because the probability of a syllable or of a word and also of the intonation of the syllable or word depends on the previous one. The coded characters are conditioned in a transmission stage known per se and therefore not illustrated in the figure, and transmitted via a first logic channel CH1 to the receiver where they are firstly preprocessed in terms of RF, and if appropriate also in accordance with the specifications of a particular mobile radio protocol by despreading, descrambling or the like in a receiver stage, which is also known and therefore omitted here, and then fed to a voice decoder 19. The further receiver-end signal processing is described further below.

[0018] The coded voice data which is made available at the output of the transmitter-end coder 17 is transmitted not only to the receiver but also immediately subjected to decoding again in a transmitter-end voice decoder 21 which functionally corresponds completely to the receiver-end voice decoder 19. A transformation into an n-dimensional state space is then carried out in a first transformation stage 23a by [means of] algorithms which are known per se. The signal which is branched at the node 11 is also subjected to a corresponding transformation in the second sub-signal path 13b in a second transformation stage 23b after said signal has been subjected to a delay, synchronized to the signal delay in the first sub-signal path 13a, in a delay stage 25. It is to be noted here that the data signal which is present at the input of the first transformation stage is a standardized voice signal which is reduced by the process of coding and subsequent decoding in the stages 17, 21, while the data signal which is present at the input of the second transformation stage 23 is still the entirety of the data signal which is merely suitably delayed. The delay time which is impressed by the delay stage 25 is controlled in adaptation to the delay in the processing chain of the stages 15, 17 and 21; in the figure it is assumed (in a somewhat simplifying fashion) that control is carried out as a function of the result of the voice recognition, i.e. on the basis of the voice recognition stage 15.

[0019] The description in the n-dimensional state space makes it possible for the individual features of the voice to be obtained in isolation as a result of the difference formation in the subtraction stage 27. Owing to the precise synchronization of the two sub-signal paths 13a, 13b, these individual features can be unambiguously assigned to the characters obtained as a result of voice recognition, and can be stored in this assignment in a transmitter-end individual-feature knowledge base 29.

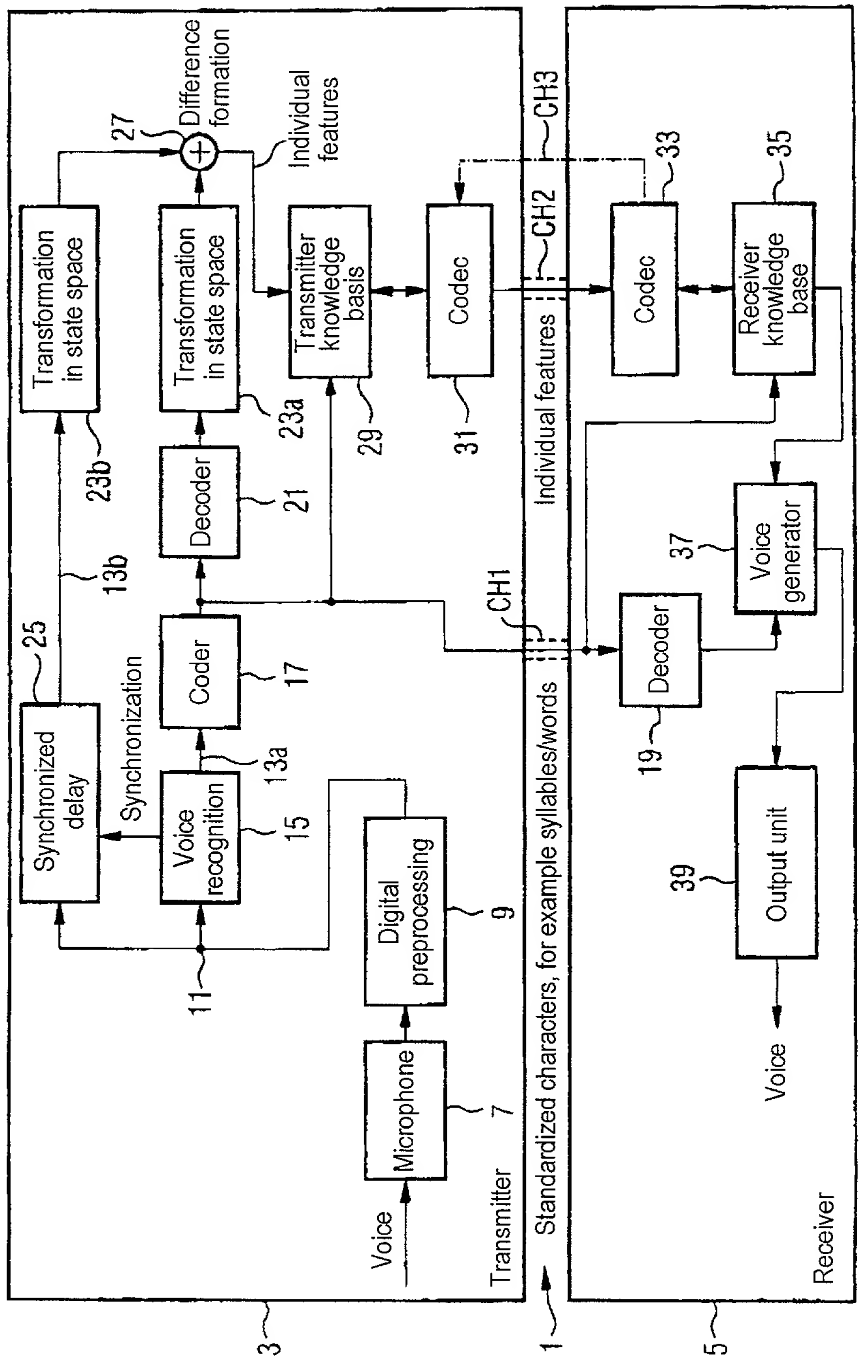
[0020] The individualization data which represents the individual features is transmitted to the receiver 5 via a separate, second logic channel CH2 at whose beginning and end there is in each case a Codec 31, 33. This data is firstly tested in the receiver using a specific control channel CH3 in a comparator and memory control stage (not shown in the figure) to determine whether or not it is already contained in a receiver-end individual-feature knowledge base 35. If this is not the case, it is stored in the receiver-end knowledge base 35 - again in assignment to the corresponding characters of the standardized voice signals (transmitted separately to the receiver 5). The receiver-end knowledge base 35 is thus to a certain degree "harmonized" with the transmitter-end knowledge base 29 in terms of its data stock so that only information relating to the individual features which is not already present in the receiver-end knowledge base 35

has to be transmitted via the separate channel. The quantity of data which is to be transmitted here can therefore be kept comparatively small.

[0021] Because the individualization data also has virtually no indispensable information value, it is transmitted with lower priority than the standardized voice signals. A transmission of this data can therefore be carried out, for example, only in the pauses during speaking or in time intervals of high word expansion. The proposed solution therefore requires a separate logic data channel but does not require any additional channel resources. The separate transmission of the individualization data from the standardized voice data in the context of the provision of a "simultaneously learning" knowledge base for the individualization data at least in the receiver (preferably in the transmitter and receiver) even makes possible a large reduction in the transmission bandwidth required despite the fact that a relatively high voice quality is achieved (dependent of course on the expenditure on the processing of the individualization features).

[0022] The individualization data which is available in the receiver-end individual-feature knowledge base 35 - again synchronized with the standardized voice signal data leaving the decoder 19 - is fed to a voice synthesis unit ("voice generator") 37 where the standardized voice signals are combined with the individualization data and an acoustic output unit 39 for converting sound is connected to its output. The method of operation of the voice synthesis unit 37 and of the output unit 39 are known per se and are therefore not explained in more detail here; a particular feature of the voice synthesis unit 37 is, however, the additional input for the individualization data and the implementation of an algorithm which is suitable for combining this individualization data with the standardized voice signals.

[0023] The embodiment of the invention is not restricted to the example outlined here but is also possible in a multiplicity of refinements and specific applications, a number of which have been already mentioned further above. <u>Thus, it will be understood that variations and modifications can be effected within the spirit and scope of the invention.</u>


Transmitter
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Microphone
7
Digital preprocessing
9
11
Voice recognition
15
Synchronized delay
25
Synchronization
13a
Coder
17
13b
Decoder
21
Transformation in state space
23a
Transformation in state space
23b
27
Difference formation
Individual features
Transmitter knowledge basis
29
Codec
31
Standardized characters, for example syllables/words
Individual features
CH1
CH2
CH3
Codec
33
Receiver knowledge base
35
Voice generator
37
Decoder
19
Output unit
39
Voice
Receiver
3
1
5

Declaration and Power of Attorney For Patent Application

Erklärung Für Patentanmeldungen Mit Vollmacht

German Language Declaration

Als nachstehend benannter Erfinder erkläre ich hiermit an Eides Statt:

dass mein Wohnsitz, meine Postanschrift, und meine Staatsangehörigkeit den im Nachstehenden nach meinem Namen aufgeführten Angaben entsprechen,

dass ich, nach bestem Wissen der ursprüngliche, erste und alleinige Erfinder (falls nachstehend nur ein Name angegeben ist) oder ein ursprünglicher, erster und Miterfinder (falls nachstehend mehrere Namen aufgeführt sind) des Gegenstandes bin, für den dieser Antrag gestellt wird und für den ein Patent beantragt wird für die Erfindung mit dem Titel:

Verfahren und Anordnung zur Sprachkodierung mittels phonetischer Dekodierung und Übertragung von Sprechermerkmalen

deren Beschreibung

(zutreffendes ankreuzen)
☐ hier beigefügt ist.
☒ am 24.05.2000 als
PCT internationale Anmeldung
PCT Anmeldungsnummer PCT/DE00/01662

eingereicht wurde und am _______
abgeändert wurde (falls tatsächlich abgeändert).

Ich bestätige hiermit, dass ich den Inhalt der obigen Patentanmeldung einschliesslich der Ansprüche durchgesehen und verstanden habe, die eventuell durch einen Zusatzantrag wie oben erwähnt abgeändert wurde.

Ich erkenne meine Pflicht zur Offenbarung irgendwelcher Informationen, die für die Prüfung der vorliegenden Anmeldung in Einklang mit Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) von Wichtigkeit sind, an.

Ich beanspruche hiermit ausländische Prioritätsvorteile gemäss Abschnitt 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 119 aller unten angegebenen Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde, und habe auch alle Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde nachstehend gekennzeichnet, die ein Anmeldedatum haben, das vor dem Anmeldedatum der Anmeldung liegt, für die Priorität beansprucht wird.

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled

Method and arrangement for speech coding, using phonetic decoding and the transmission of speech characteristics

the specification of which

(check one)
☐ is attached hereto.
☒ was filed on 24.05.2000 as
PCT international application
PCT Application No. PCT/DE00/01662
and was amended on ___________
(if applicable)

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, §119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

IDNR: 2590 / V: 99-1.00 / B:Val

Form PTO-FB-240 (8-83)

Patent and Trademark Office-U.S. DEPARTMENT OF COMMERCE

German Language Declaration

Prior foreign appplications
Priorität beansprucht

Priority Claimed

19925264.5 (Number) (Nummer)	DE (Country) (Land)	01.06.1999 (Day Month Year Filed) (Tag Monat Jahr eingereicht)	☒ Yes Ja	☐ No Nein
(Number) (Nummer)	- (Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	☐ Yes Ja	☐ No Nein
(Number) (Nummer)	-(Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	☐ Yes Ja	☐ No Nein

Ich beanspruche hiermit gemäss Absatz 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 120, den Vorzug aller unten aufgeführten Anmeldungen und falls der Gegenstand aus jedem Anspruch dieser Anmeldung nicht in einer früheren amerikanischen Patentanmeldung laut dem ersten Paragraphen des Absatzes 35 der Zivilprozeßordnung der Vereinigten Staaten, Paragraph 122 offenbart ist, erkenne ich gemäss Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) meine Pflicht zur Offenbarung von Informationen an, die zwischen dem Anmeldedatum der früheren Anmeldung und dem nationalen oder PCT internationalen Anmeldedatum dieser Anmeldung bekannt geworden sind.

I hereby claim the benefit under Title 35. United States Code. §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §122, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occured between the filing date of the prior application and the national or PCT international filing date of this application.

PCT/DE00/01662 (Application Serial No.) (Anmeldeseriennummer)	24.05.2000 (Filing Date D, M, Y) (Anmeldedatum T, M, J)	anhängig (Status) (patentiert, anhängig, aufgegeben)	pending (Status) (patented, pending, abandoned)
(Application Serial No.) (Anmeldeseriennummer)	(Filing Date D,M,Y) (Anmeldedatum T, M; J)	(Status) (patentiert, anhängig, aufgeben)	(Status) (patented, pending, abandoned)

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

German Language Declaration

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POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and transact all business in the Patent and Trademark Office connected therewith. *(list name and registration number)*

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